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INT CL[6] H03M 7/30, H04B 1/66, H04H 7/00, H04N 5/60 7/52
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(54) Abstract Title

Compression decoding and re-encoding

(57) An auxiliary data signal is derived from a received compression encoded audio signal during decoding of the signal and communicated with the decoded audio signal for use in re-encoding the signal. The auxiliary data signal may be communicated integrally with the audio signal, for example in the least significant bits, or may follow a different path. The provision of the auxiliary data signal may enable more transparent decoding and re-coding processes to take place, by allowing the coding decisions to match the coding decisions originally used; this can alleviate problems with quality reduction arising from cascaded decoding and re-coding processes.

Application is to digital broadcasting in the studio environment with mixing, fading, etc.

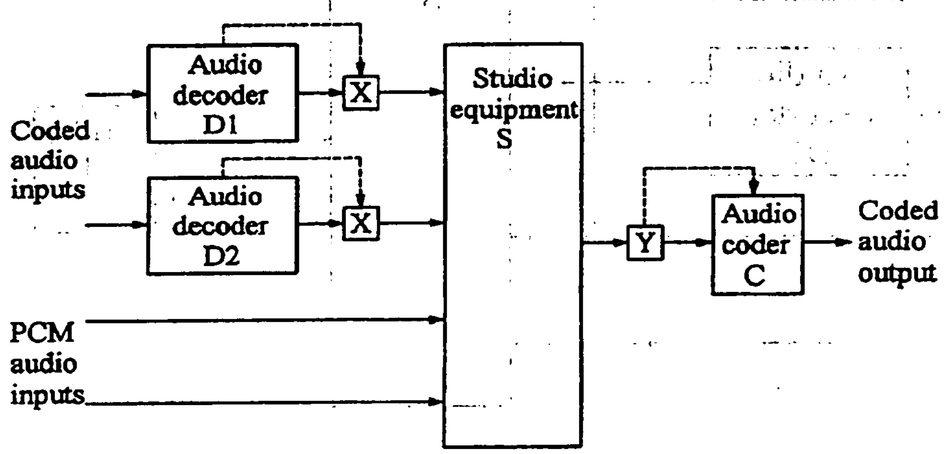

Fig. 1

At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

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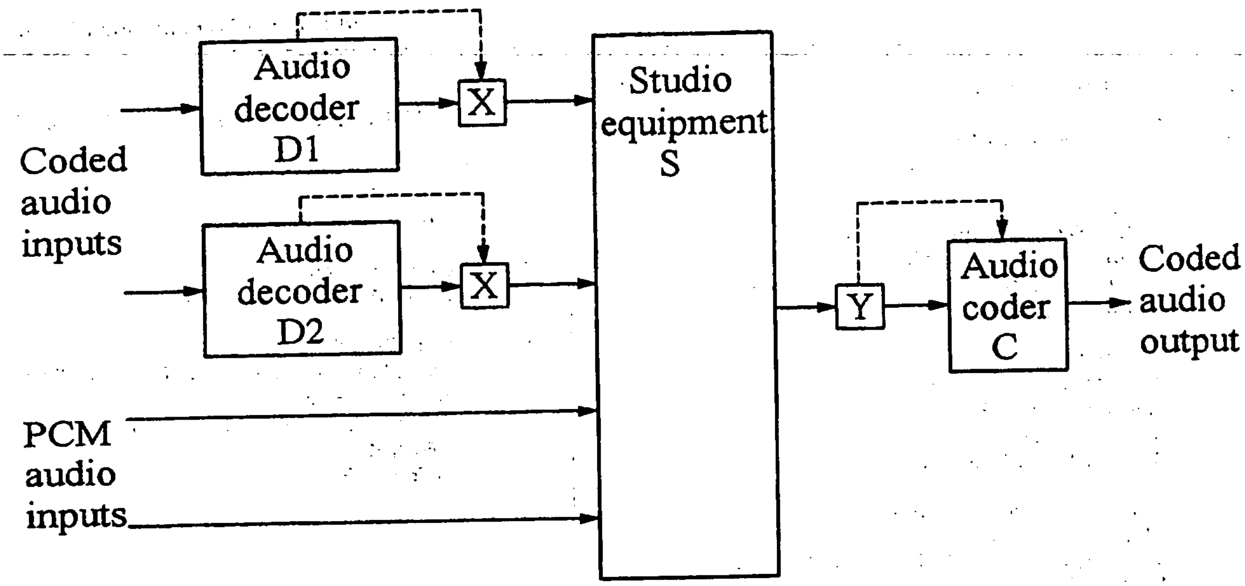
Audio
decoder
D1
X
Studio
equipment
S
Coded
audio
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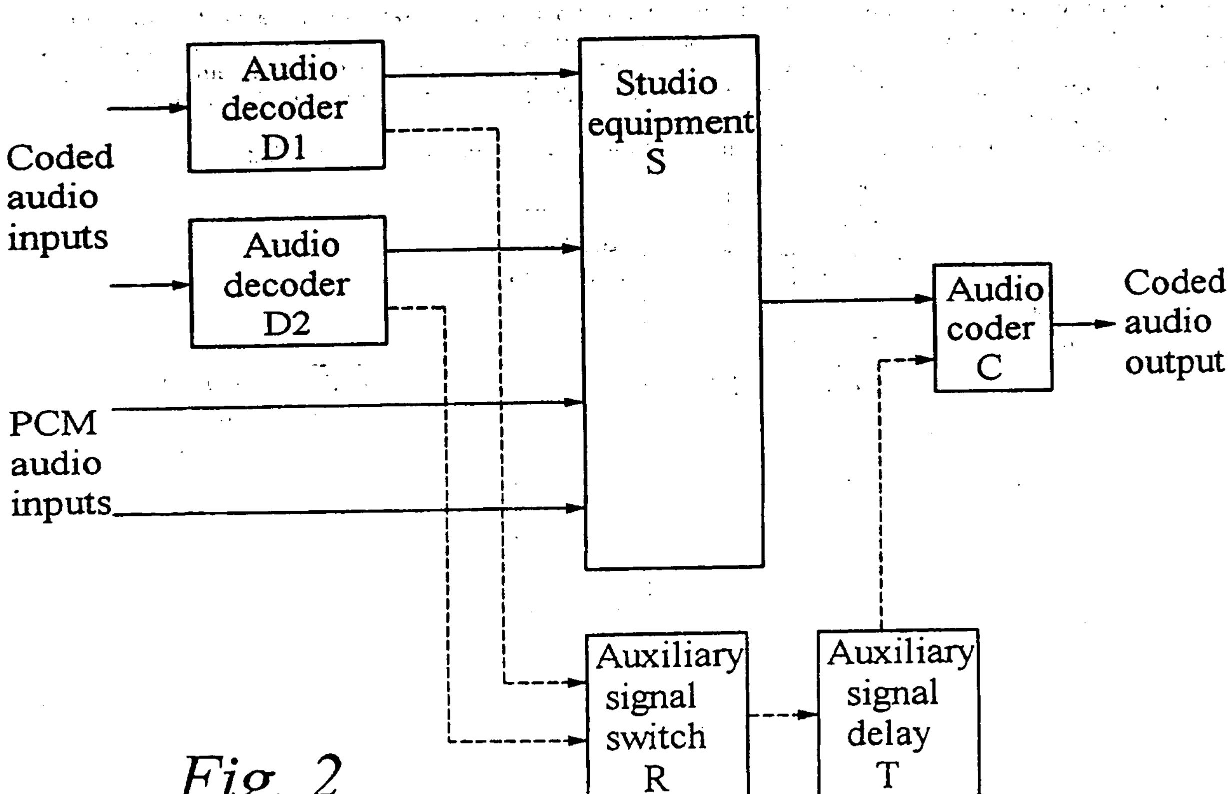
Audio
decoder
D2
X
Y
Audio
coder
C
Coded
audio
output
PCM
audio
inputs

Fig. 1


Audio
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D1
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Fig. 2

AUDIO COMPRESSION

This invention relates to compressed, that is to say data–reduced or bit–rate reduced, digital audio signals.

The invention is applicable to a wide range of digital audio compression techniques; an important example is the so–called "MPEG Audio" coding, defined in ISO/IEC standards IS 11172–3 and IS 13818–3.

In digital broadcasting, certain operations can be performed only on decoded audio signals. There is accordingly a requirement for compression decoding and re–encoding in the studio environment. It is of course desirable that these cascaded decoding and re–encoding processes should involve minimal reduction in quality. Studio operations such as mixing may be conducted on a digital PCM signal, although sometimes there will be a requirement for conversion of the PCM signal to analogue form. In the discussions that follow, attention will be focused on the use of a decoded audio signal in PCM format although it should be remembered that the invention also encompasses the use of decoded analogue signals in analogue form. It will further be appreciated that whilst the digital broadcasting studio environment conveniently exemplifies the present invention, the invention is applicable to other uses of compressed audio signals.

It is an object of the present invention, in one aspect, to provide improved digital audio signal processing which enables re–encoding of a compression decoded audio signal with minimal reduction in quality

Accordingly, the present invention consists in one aspect in a method of audio signal processing, comprising the steps of receiving a compression encoded audio signal; compression decoding the encoded audio signal; deriving an auxiliary data signal; communicating the auxiliary data signal with the decoded audio signal and re–encoding the decoded audio signal utilising information from the auxiliary data signal.

Preferably, the auxiliary data signal comprises essentially the encoded audio signal.

In one form of the invention, the auxiliary data signal is combined with the decoded audio signal for communication along the same signal path as the decoded audio signal.

The invention will now be described by way of example with reference to the accompanying drawings in which:–

Fig. 1 is a block diagram of a digital broadcasting studio installation utilising an embodiment of the present invention; and

Fig. 2 is a block diagram of similar form illustrating a second embodiment of the present invention.

Referring to Fig. 1, a coded audio bit–stream enters the decoder (D1) at the top left and the decoder produces a linear PCM audio signal, typically in the form of an ITU–R Rec. 647 ("AES/EBU") bitstream, although other forms of PCM signal may be used. The PCM signal is connected to the studio equipment (S) which may provide such facilities as fading, mixing or switching. This connection is made via an insertion unit (X) which combines the auxiliary data signal with the PCM audio signal. Other audio sources are connected to the studio equipment; these are in the form of PCM signals, but some or all of them may previously have been coded, and those decoded locally may be accompanied by auxiliary data signals (e.g. the PCM signal from Decoder D2). The output of the studio equipment is applied to the input of the coder (C) via a signal splitter unit (Y) which separates the auxiliary data from the PCM signal. The output of the coder is a coded (i.e. digitally compressed) audio signal. In Fig. 1, the PCM signal path is represented by the solid line connecting the decoder and coder via the studio equipment. If just a PCM signal arrives at the coder (i.e. the auxiliary data signal is not present) the latter has to perform an independent re–coding process. This introduces impairments in the form of coding artifacts into the signal (in the case of a PCM signal which has previously been coded, but without the auxiliary signal, these artifacts are additional to those present as the result of the earlier coding).

In the example of an MPEG audio signal, the most important

information to carry with the signal is the positions of the coded audio frame boundaries. These frames are 24ms long when the sampling frequency is 48 kHz.

The build up of impairments can be completely eliminated by avoiding decoding and re–coding wherever possible. For example, if enough of the original coded audio signal is conveyed to the coder, as the auxiliary data signal, the coded audio signal can be reconstituted and substituted for the decoded and re–coded signal. This would require that the studio equipment pass the PCM signal transparently, and that the coded bitstreams to be switched or mixed are frame aligned, or can be brought into frame alignment. Frame aligning can give rise to problems with audio/visual synchronisation ("lip sync") in applications such as television where video is associated with the audio.

Alternatively, if the auxiliary data signal indicates to the coder the positions in the PCM bitstream of the frame boundaries of the original coded signal, it is possible to minimise any impairment introduced on re–coding if the original groups of audio samples which formed blocks of coded data (e.g. subband filter blocks or blocks of samples with the same scalefactor) are kept together to form equivalent blocks in the re–coded signal. This does not require frame alignment of coded bitstreams within the studio area, but it does require alignment of the appropriate data blocks within the bitstreams. Such alignment can be effected by the introduction of relatively short delays, which do not significantly affect audio/video synchronisation. Further reductions in the impairment on re–coding may be made if information on the quantisation of the audio in the coded bitstream is conveyed to the coder (C).

A further possibility is to move frame boundaries in the incoming coded bitstreams, whilst preserving the original blocks of coded data, to bring the frames closer to alignment. Relatively short delays can then be used to effect frame alignment by "fine tuning" the timing of the signals. Frame aligning the coded bitstreams in this way, at a point where the entire incoming coded audio signal is available will minimize further impairment of

the audio, and re–coding will take place with the repositioned frame boundaries.

If the frame boundaries are repositioned in such a way as to preserve the original block of samples with the same scale factor, only a partial decoding operation is needed. This technique is particularly suited to the editing of bit–rate reduced digital signals because full decoding and re–encoding can be eliminated.

In the case where the studio is receiving MPEG audio coded signals in the form of packetised elementary streams (PES), buffer stores in the decoders are used to ensure that the audio signals are correctly timed to a local clock and (if appropriate) to associated video signals, using a programme clock reference (PCR) and presentation time stamps (PTS) within signals. The relatively small adjustments to signal timing needed to align blocks within coded bitstreams entering the studio with the blocks formed by the re–encoding process in the coder (C) may be made either by making some adjustment to the timing in the decoders (D1, D2 etc.) or by introducing delays into the PCM signal paths.

In the arrangement of Fig. 1, the auxiliary data takes the same path as the PCM signal through the studio equipment, and is combined with the PCM audio in such a way that it has the minimal effect upon the audio. It is routed with the audio, and if the path is not transparent (e.g. because of fading or mixing) the modification of the auxiliary signal is detected in the coder, and re–coding of the audio proceeds independently of the auxiliary signal. If the path is transparent, the unmodified auxiliary signal facilitates the substitution of the re–coded PCM signal by the original coded signal, or re–coding with the data blocks of the re–coded signal reproducing the blocks of the original signal as closely as possible, as described above. The dotted line of Fig. 1 represents the path taken by the auxiliary data.

Any modification of the signal and associated auxiliary data is detected by appropriate examination of the auxiliary data. For example, the auxiliary data may be accompanied by error–detecting cyclic redundancy check bits associated with the auxiliary data for each coded audio frame.

Audio signals which have not previously been coded will not be accompanied by any auxiliary data and will be impaired by the coding artifacts introduced by first–time coding when coded by the coder (C). Signals which have previously been coded but for which no auxiliary data is available will be impaired by additional coding artifacts when re–coded by the coder (C).

Referring to Fig. 2, the PCM audio signal takes the same path through the studio equipment from the decoder (D1) to the coder (C) via the studio equipment (S). However, in this arrangement, the auxiliary data signal is not combined with the PCM audio but is routed separately. This arrangement has the advantage that the auxiliary data is not combined with the PCM audio, and there is no risk of audible changes to the signal as a result. This might be important, for example, if the studio equipment has only a limited resolution in terms of the audio sample word–length. Furthermore, the auxiliary data is not modified by fading or mixing. There are disadvantages in that the auxiliary signal needs to be delayed to keep it time–aligned with the PCM audio passing through the studio equipment (5), and switching is necessary in the auxiliary data path so that the correct auxiliary data is always presented to the coder (C) with the associated PCM signal. As in the arrangement of Fig. 1, the coder needs to perform re–coding independently of the auxiliary signal at times when the path through the studio equipment (S) is not transparent. One way of ensuring that this happens is for the switch (R) which routes the auxiliary signals the coder to suppress all such signals when independent re–coding is necessary. Another way would be to add a subsidiary auxiliary data signal to the audio passing through the studio equipment (S) which would enable detection of non–transparent processing. This might be, for example, a known pseudorandom binary sequence (prbs) or some form of cyclic redundancy check data on some or all of the audio data.

In Fig. 2, the delay (T) required in the auxiliary data path should be constant, and may be determined by means of suitable tests. However, incoming MPEG audio coded bitstreams in PES form contain PTS, as

mentioned previously, and PCM audio signals can carry time information (e.g. the time codes in the ITU-R Rec. 647 signal) which may comprise, or be derived from, the incoming PTS. If the auxiliary signal contains the same information, or the PTS itself, the initial setting of the delay (T) and the subsequent verification of the amount of delay may be performed automatically.

Examples of signals that could comprise the auxiliary data are:

1. The coded audio signal at the input to the decoder (D1, D2, etc.). This contains not only audio-related data and the PTS but also certain auxiliary information such as programme-associated data (PAD), which may need to be copied into the coded signal at the output from the studio area, and error protection. Depending upon the circumstances, such a signal would enable the coder (C) to substitute the original coded signal for the re-coded PCM signal, or to re-code the PCM signal with blocks of audio data resembling closely the blocks within the original coded signal, as described above. Conveying the coded audio signal to the coder provides the widest range of options for re-coding with minimal additional impairment of the audio.

2. The coded audio samples at the input to the decoder minus the quantised audio samples (which can be re-created identically from the PCM audio signal). This is a signal in which the positions of the frame boundaries of the original coded signal are indicated relative to the linear audio samples in the PCM signal, and from which the positions of the blocks of data within the frames may be deduced, together with information on the allocation of bits to the various components of the coded signal (sometimes known as "bit-allocation data"), scale factors, block lengths (in coding schemes where this is relevant), the PTS, and any other data relevant to the coding system in use.

3. A signal similar to that described in "2" above, but containing a subset of the information described (e.g. just the positions of the frame boundaries).

Ways in which the auxiliary data signal might be transported with the PCM audio are:

1. In the auxiliary sample bits of the ITU-R Rec. 647 bitstream. At the studio standard sampling frequency of 48 kHz, a total bit rate of 384 kbit/s is available in the auxiliary sample bits of both "X" and "Y" subframes. This method is ideal for conveying the auxiliary data between different items of equipment but there is some uncertainty concerning the way in which studio equipment might treat these auxiliary sample bits. For example, the studio equipment may not route these bits through to the output with the PCM audio, or it may not delay these bits by the same amount as the PCM audio. In either case, some modification of the studio equipment, or of the environment around it, may be necessary.

2. In the least significant bits (l.s.b.) of the PCM audio sample words of the ITU-R Rec. 647 bitstream. Depending upon the resolution of the studio equipment these may the same as the auxiliary sample bits (these are the l.s.b if the Rec. 647 signal is configured to carry 24-bit audio sample words) or the least significant bits within the part of the subframe reserved for 20-bit audio sample words (these are the same bits that carry the 20 most significant bits of 24-bit sample words). Carrying the auxiliary data in the l.s.b. of the audio sample words overcomes the problems of routing within the studio equipment and care will be taken to ensure that the auxiliary data signal is inaudible. The studio equipment needs to be transparent to audio sample words of at least 20 bits. If necessary, the audibility of the auxiliary data signal could be reduced by scrambling (e.g. by the

modulo-2 addition of a pseudorandom binary sequence, or the use of a self-synchronising scrambler). Alternatively, it could be removed altogether by truncating the audio sample words to the appropriate length (i.e. to exclude the auxiliary data).

3. In the user data bits of the ITU-R Rec. 647 bitstream. Taking the user data bits from both "X" and "Y" subframes provides a channel with a bit rate of only 96 kbit/s. In many applications this is unlikely to be sufficient to carry the complete coded audio signal. It would be sufficient to signal the positions of frame boundaries, and to carry some other information extracted from the coded audio. With this method there is uncertainty concerning the way in which studio equipment might treat the user data.

4. In the upper part of the audio spectrum, at frequencies higher than those of the audible components of the signal. For this purpose, the PCM audio signal would be low-pass filtered, and the coded auxiliary data signal added above the passband occupied by the audible signal. A particularly ingenious way of doing this, when the studio area is receiving MPEG audio coded signals, would be to use an MPEG analysis subband filterbank with the reciprocal synthesis filterbank at the insertion units (X) in Fig. 1. At 48 kHz sampling frequency, the audio passband extends almost up to 24 kHz. In MPEG audio coding this passband is divided into 32 equally-spaced subbands, each with a bandwidth of 750 Hz. The upper five subbands are not used, and the audio is thus effectively low-pass filtered to 20.25 Khz. The auxiliary data could be inserted into the upper subbands, and would be carried in the upper part of the spectrum of the PCM audio signal, to be extracted by another MPEG analysis filterbank at the splitter (Y) shown in Fig. 1. The PCM signal applied to the coder (C) would not need further filtering to remove the auxiliary data, as this would happen in the analysis filterbank in the

coder itself.

5. The auxiliary signal might be a low–level known pseudo random binary sequence (prbs) added to the audio. The prbs would be synchronised in some way with the audio frame boundaries and may be modulated with additional data where possible. It is also possible to subtract the prbs from the data prior to final transmission or monitoring.

It has been explained that under certain circumstances it is appropriate to perform partial decoding and re–encoding. In the appended claims, the terms decoding and re–encoding should be taken as including partial decoding and re–encoding, respectively.

CLAIMS

1. A method of audio signal processing, comprising the steps of receiving a compression encoded audio signal; compression decoding the encoded audio signal; deriving an auxiliary data signal; communicating the auxiliary data signal with the decoded audio signal and re-encoding the decoded audio signal utilising information from the auxiliary data signal.

2. A method according to Claim 1, wherein the auxiliary data signal comprises essentially the encoded audio signal.

3. A method according to Claim 1, wherein the auxiliary data signal comprises audio-related data from the encoded audio signal.

4. A method according to Claim 3, wherein the auxiliary data signal comprises time information from the encoded audio signal.

5. A method according to Claim 4, wherein the auxiliary data signal further comprises ancillary information, such as programme-associated data, from the encoded audio signal.

6. A method of audio signal processing, comprising the steps of receiving a compression encoded audio signal; compression decoding the encoded audio signal; deriving an auxiliary data signal indicative of the analysis and quantisation employed for the encoded audio signal; communicating the auxiliary data signal with the decoded audio signal and re-encoding the decoded audio signal utilising information from the auxiliary data signal such that the re-encoded audio signal employs the same analysis and quantisation as the encoded audio signal.

7. A method according to Claim 6, wherein the analysis comprises application of sub-band filter bank.

8. A method according to Claim 7, wherein the auxiliary data signal is indicative of the analysis into sub-bands and the quantisation within each sub-band employed for the encoded audio signal.

9. A method according to any one of the preceding claims, wherein the encoded audio signal is an MPEG audio coded signal.

10. A method according to Claim 9, wherein the auxiliary data signal contains information relating to one or more of: the position of audio frame boundaries in the audio signal; scale factors for the blocks of sub-band samples within each audio frame; bit allocation data for the audio frame.

11. A method according to any one of the preceding claims, wherein the auxiliary data signal is combined with the decoded audio signal for communication along the same signal path as the decoded audio signal.

12. A method according to Claim 11, wherein the auxiliary data signal is formatted to enable an integrity check prior to use of the auxiliary data signal in a re-encoding process, to ensure transparent communication of the auxiliary data signal along the decoded audio signal path.

13. A method according to Claim 11, wherein the auxiliary data signal is carried in the least significant bits of a digital decoded audio signal.

14. A method according to Claim 11, wherein the auxiliary data signal is carried as user data bits in a recognized digital interface format such as ITU-R Rec. 647.

15. A method according to Claim 11, wherein the auxiliary data signal is carried in the upper part of the audio spectrum.

16. A method according to Claim 15, wherein the auxiliary data signal is carried in higher frequencies associated with sub–bands unused in the compression encoding.

17. A method according to Claim 16, in which MPEG audio coding is employed, wherein a filter arrangement analogous to the MPEG analysis sub–band filter arrangement and its reciprocal, is employed for insertion of the auxiliary data signal into the decoded audio signal.

18. A method according to any one of Claims 1 to 10, wherein the auxiliary data signal is carried in a separate path to the decoded audio signal.

19. A method according to Claim 18, wherein the auxiliary data signal path is disabled in the event of processing in the decoded audio signal preventing sensible use of information from the auxiliary data signal in re-encoding.

20. A method according to Claim 19, wherein a tell–tale is added to the decoded audio signal to indicate such processing.

Application No:	GB 9701616.6	**Examiner:**	Keith Williams
Claims searched:	1-20	**Date of search:**	30 June 1997

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.O): H4P (PDCFX); H4R (RPBE, RPCX, RPNR, RPX, RSX)

Int Cl (Ed.6): H03M 7/30; H04B 1/66; H04H 7/00; H04N 5/60, 7/52

Other: online WPI

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
A	EP 0640909 A1	Texas Instruments Inc. - see columns 1-3 and column 12, lines 33-49 (and equivalent US 5568495)	
A	US 5185800	Centre National D'Etudes des Telecom. - see column 9, lines 33-37 (and equivalent EP 0423050 A1)	

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
		E	Patent document published on or after, but with priority date earlier than, the filing date of this application.
&	Member of the same patent family		